REMARKS

Another set of formal drawings are being submitted herewith. The specification has been amended to correct grammatical mistakes. No new subject matter has been added. Claims 1-17 remain in the subject application. Dependent claims 18-20 are added to the subject application. Claims 1, 14 and 17 have been amended, as recited hereinabove.

Claims 1-13 and 17 have been rejected under 35 U.S.C. 103(a) as being unpatentable over Vargo et al. (U.S. Patent No. 6,356,545) in view of Blomfield-Brown et al. (U.S. Patent No. 6,292,840).

It is stated on page 2, paragraph 1 of the office action that with respect to claims 1 and 17, the limitations recited in claim 1, a "DSP module responsive to an analog signal from one of the telephone devices … and further operative to packetize digital telephone signal for transmission to a …" are inherently addressed by Vargo. Applicant respectfully disagrees with the same, as there is no teaching of such a DSP module by Vargo. Additionally, the DSP module of the claimed invention is discussed on page 7 , lines 22-25 of the subject application as including "a number of DSP chips (or integrated circuits), which are special purpose processors for efficiently executing mathematical operation, such as multiply and add operations, in one clock cycle." Again, such a DSP module does not appear to be taught by Vargo et al..

Furthermore, claims 1 and 17, as amended recite "renegotiating a second type of codec", which is not taught by Vargo et al. This is not taught either by Vargo et al. or by Blomfield-Brown et al. Furthermore, neither reference teaches repeated negotiation of the type of codec employed to "dynamically change compression techniques to adjust for network usage thereby …", as recited by the amended claims.

The combination of Vargo et al. and Blomfield-Brown et al. is objected thereto as neither suggests or hints at the teachings of the other in a way as to render the claimed invention obvious to one of ordinary skill in the art. While, in hindsight, the combination of these two references may appear to be obvious to one of ordinary skill in the art after the

making of the claimed invention, the same is simply not true at the time of the making of the claimed invention.

It is believed that claims 1-13 and 17, as amended, are patentable over Vargo et al. in view of Blomfield-Brown et al. and therefore patentable. It is further believed that all claims depending therefrom are also patentable. Reconsideration and allowance of the same is respectfully requested.

Claims 14-16 have been rejected under 35 U.S.C. 103(a) as being unpatentable over Schuster et al. (U.S. Patent No. 6,483,600) in view of Blomfield-Brown (U.S. Patent No. 5,625,678).

Schuster et al. does not teach or disclose transmission of both voice and fax signals, as claimed in the amended claim 14. For example, Schuster et al. does not teach "a DSP module for carrying a user-initiated telephone conversation ..." In fact, system 10 of Schuster et al. is a "data network facsimile system" and no disclosure is found of the combination of voice and fax. Blomfield-Brown et al. discloses a method and system for allowing multiple application programs to communicate in the context of a switched voice and data communication (see Blomfield-Brown et al.: Abstract). This is not the teachings of the claimed invention as the latter is a "router device for use in a communication system ..."

The combination of Schuster et al. and Blomfield-Brown et al. is objected thereto as there is no teaching, suggestion or hint of one by the other.

It is therefore believed that claim 14, as amended, is patentable over Schuster et al. in view of Blomfield-Brown et al. Claims 15 and 16 depending from claim 14 are therefore believed to be patentable. Reconsideration and allowance of claims 14-16 is hereby requested.

Consideration and allowance of claims 18-20 is hereby respectfully requested.

Applicants submit that the application is now in condition for allowance and an early notice thereof is requested. Should any further amendment be required prior to passing the application to issue, the Examiner is respectfully invited to contact the undersigned by telephone at the number set out below.

Respectfully submitted,

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I hereby certify that this correspondence with all attachments is being deposited with the U.S. Postal Service as first class mail in an envelope addressed to: Box No Fee Amendment, Assistant Commissioner for Patents, Washington, D.C. 20231 on January 29, 2003 by Erika Villafana.

Erika Villafana

VERSION WITH MARKINGS TO SHOW CHANGES MADE

In the Specification:

The specification has been amended as follows:

The paragraph starting on page 1, line 8 and ending at page 1, line 13 has been replaced with the following paragraph:

--The present invention relates generally to dynamically changing compression techniques used for transmission of voice information over a packet switching environment and for transmission of fax information on a telephone line, in real-time, the telephone line being also used for transmission of voice information over a packet switching network environment and particularly [to] for loading an overlay compression layer during a telephone conversation upon user initiation or automatically.—

The paragraph starting on page 3, at line 17 and ending at page 3, line 24 has been replaced with the following paragraph:

--A problem that arises with respect to prior art systems similar to Fig. 2 is that only one type of compression algorithm is employed for [any] the duration of a phone call. That is, compression algorithms can not be changed during the phone conversation. The use of only one type of compression technique prevents compensation for variable factors, such as varying packet sizes. Therefore, the need arises for the use of dynamically changing compression techniques, either manually or automatically for transmission of voice-over-IP allowing for adjustments to be made in accordance with varying network sage or bandwidth thereby making optimal use of network capacity and throughput.—

The paragraph starting on page 13, at line 3 and ending at page 13, line 7 has been replaced with the following paragraph:

--It should be noted that a similar flow chart can be used to perform the opposite. Consider the case where a type of codec is being executed that compensates for either packet loss or pursuant to a user request, if, after sometime, the quality of information transfer over the network improves such that more bandwidth becomes available, another codec that results in [a] larger packet sizes but a higher quality of transmission may be negotiated therefore and loaded.--

In the Claims:

The following claims have been amended:

1. (Twice Amended) A router device for use in a communication system having at least two telephone devices in communications with each other for transferring voice information therebetween through a packet switching network, the router device being coupled between one of the telephone devices and the packet switching network and for performing one of a plurality of types of compression/decompression (codec) operation on information being transferred between the telephone devices comprising:

a Digital Signal Processor (DSP) module responsive to an analog telephone signal from one of the telephone devices and operative to convert the analog telephone signal to a digital telephone signal and further operative to packetize the digital telephone signal for transmission to a remotely-located router device, the router device and the remotely-located device initially negotiating to utilize a first type of codec, the DSP module for renegotiating the use of a second type of codec and switching from using said first type of codec to using [a] said second type of codec upon detection of degradation in the quality of the voice information,

wherein the type of codec being utilized is repeatedly renegotiated to dynamically change compression techniques to adjust for network usage thereby optimizing the use of network capacity and throughput and further wherein switching between the codecs is performed while a conversation is taking place between the two telephone devices yet avoiding substantial disturbance to users of the telephone devices.

14. (Once Amended) A router device for use in a communication system having a first telephone device for causing the transmission of voice conversations and a first fax machine coupled to the router device, the router device responsive to telephone signals, carrying voice conversations, generated by the first telephone device and fax signals generated by the first fax machine and operative to transfer digital information, through a packet switching network, to a remotely-located router coupled to a second telephone device for receiving the voice conversations and a second fax machine comprising[;]:

a DSP module for carrying a user-initiated telephone conversation on a telephone line connecting the first telephone device and the second telephone device through the packet switching network, the DSP module further responsive to analog fax signals form the first fax machine and further operative to convert the analog fax signals to digital fax signals and to packetized the digital fax signals for transmission, through the packet switching network, to the second fax machine,

wherein the fax transmission from the first fax machine to the second fax machine takes place on the telephone line causing a temporary interruption to the telephone conversation thereby avoiding the need for telephone connection to be disconnected prior to the fax transmission.

17. (Twice Amended) A method for use in a communication system having at least two telephone devices in communications with each other for transferring voice information therebetween through a packet switching network, the router device being coupled between one of the telephone devices and the packet switching network and for performing one of a plurality of types of compression/decompression (codec) operation on information being transferred between the telephone devices comprising:

receiving an analog telephone signal through a telephone connection from one of the telephone devices;

converting the analog telephone signal to a digital telephone signal;

separating information carried on the digital telephone signal into packets of information;

initially negotiating a first type of codec for communication between the telephone devices;

using a first type of codec for transferring the packets of information between the two telephone devices through the packet switching network; [and]

renegotiating the use of a second type of codec; and

switching to using [a] said second type of codec upon detection of degradation in the quality of the voice information during the course of the telephone connection.

The following claims have been added:

18. A router device as recited in claim 1 wherein the codec negotiation is performed pursuant to the H.245 protocol.

19. A router device as recited in claim 1 wherein the first type of codec utilizes a compression/decompression algorithm defined by any one of the standards: G.711, G726, G729 or G723.1 and the second type of codec utilizes a compression/decompression algorithm defined by any one of the standards: G.711, G726, G729 or G723.1.

20. A router device as recited in claim 14 wherein the connections are established pursuant to the H.225 protocol.